PATENT APPLICATION

K. Ward
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In re the Application of

Lars ARKNÆS-PEDERSEN

Application No.: New U.S. Patent Application

Filed: April 1, 1999 Docket No.: 103176

For: A METHOD AND AN APPARATUS FOR PROCESSING AN AUSCULTATION SIGNAL

JC518 U.S. PTO
09/283585
04/01/99

CLAIM FOR PRIORITY

Assistant Commissioner for Patents
Washington, D.C. 20231

Sir:

The benefit of the filing date of the following prior foreign application filed in the following foreign country is hereby requested for the above-identified patent application and the priority provided in 35 U.S.C. §119 is hereby claimed:

Danish Patent Application No. 0515/98 filed April 8, 1998

In support of this claim, a certified copy of said original foreign application:

__X__ is filed herewith.

_____ was filed on _____ in Parent Application No. _____ filed _____.

It is requested that the file of this application be marked to indicate that the requirements of 35 U.S.C. §119 have been fulfilled and that the Patent and Trademark Office kindly acknowledge receipt of this document.

Respectfully submitted,

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Patent application No.: 0515/98

Date of filing: 08 Apr 1998

Applicant: Bang & Olufsen Technology A/S, Bødkervej 2, DK-7600 Struer, DK

This is to certify the correctness of the following information:

The attached photocopy is a true copy of the following document:

- The specification, claims, abstract and drawings as filed with the application on the filing date indicated above.

Patentdirektoratet · Kongeriget Danmark

Erhvervsministeriet
Patentdirektoratet

TAASTRUP 30 Mar 1999

Heidi Meldgaard Nielsen

Heidi Meldgaard Nielsen
Assistent

A method and an apparatus for processing a signal.

Background of the invention.

The present invention relates to a method of processing a signal representing an input sound signal, said signal being divided in time into a plurality of signal segments, each having an individual duration of time, said signal segments being processed into an output signal of successive signal segments, said signal segments being processed in such a way that at least one, preferably all signal segments are repeated immediately and successively at least once in said output signal.

Moreover, the present invention relates to a sound monitoring apparatus, and in particular to electronic stethoscopes for use by physicians and especially for those suitable for use in cardiology.

Field of the art.

Through the recent years physicians have been provided with an impressive arsenal of instrumentation for the diagnosis of cardiovascular diseases. Such an instrument is the well known stethoscope used to detect sounds originating from the heart and adjacent large vessels. Sound monitoring of the heart, or auscultation in general, is an important aspect in the evaluation of the physical condition of an individual, and is particularly important in the diagnosis of certain pathological conditions which manifest themselves by abnormal sounds.

When using a normal bifurcated stethoscope with binaural earpieces and a bell or diaphragm for receiving the sound signal, it is difficult to distinguish the sound elements

in fast beating hearts, e.g. infants, but also when auscultating patients with a "normal" heart rate, it can be difficult to observe split heart sounds or a weak murmur located near a primary heart sound.

Today it is possible to process the information residing in the auscultation signal electronically by using knowledge obtained by clinical research. Electronic stethoscopes make it possible to modify the physiological signal, but the approaches are mostly based on changes of the frequency components in the signal, which makes it difficult for a physician, trained in the use of the conventional stethoscopes, to recognize the signal.

This leads to the goal of creating a stethoscope or an apparatus for auscultation in general that makes it easier for the pathologist to distinguish between the different sound elements in even fast heart sounds. Since the pathologists partly base their diagnosis on the heart sounds, it is of great importance that an exact reproduction of the sound elements in the sound signal is performed, meaning that there should be no change in the pitch of the signal and no dissonance should be added as a result of the reproduction algorithm. If either distortion or change of pitch is present, it could lead to a wrong interpretation of the heart sounds, resulting in incorrect diagnoses by the physicians.

US patent No. 4,528,689 discloses the idea of a method for artificially slowing down an analyzed sound signal. It is done by first low pass filtering the sound signal from the heart and then splitting the signal, which varies cyclically from zero crossing to zero crossing, into a number of cycles, and each cycle is repeated succes-

sively. These repetitions of half-periods of a sound representing signal result in a slow version of the original sound having the original pitch.

A serious disadvantage of the above-mentioned method is that the resulting signal provides echoes in such a way that a listener may obtain confused results while listening to the generated slow signal. Appearances of echoes might result in a wrong interpretation of some sound elements, and these sound elements are often of vital importance. When identifying a heart disease, this method might lead to incorrect diagnoses. Further the method introduces click sounds at the points where successive cycles are pasted together. Dissonance as click sounds might also lead to disturbance of the auscultation signal and even wrong diagnoses. Apparently, the method has found very little commercial use, if any.

It is an object of the invention to provide a method which will be able to slow down sound signals especially sound signals representing auscultation signals like heart sounds, while still obtaining the original pitch with a minimum of echo in the resulting slowed down signal.

SUMMARY OF THE INVENTION

When, as stated in claim 1, a method of processing a signal representing an input sound signal, said signal being divided in time into a plurality of signal segments each having an individual duration of time, said signal segments being processed into an output signal of successive signal segments, said signal segments being processed in such a way that at least one, preferably all signal seg-

ments are repeated immediately and successively at least once in said output signal

wherein

each signal segment is established in such a way that the duration of time of substantially all the signal segments is less than 60 ms, a very advantageous signal processing method is obtained, as the input signal may be reproduced as a slow-speed signal having substantially the same signal components as the input signal. Keeping the signal segments below 60 ms, the repetition of sound segments will not be perceived as echoes by the physician.

It will thus be possible for a listener to distinguish between the different sounds of the generated output signal, as the audio signals generated according to the invention are ideally free of any perception of echo or significant distortion. This possibility of eliminating the echo perception for a listener makes the invention a unique and essential part of any analysis tool for supporting signal analyzing based on subjective recorded and processed audio signals. Using this invention the physician will not be able to distinct between a "real" signal and a processed signal, as the processed signal will be perceived as a "real" signal.

The basic results achieved according to the invention are based upon experimental experience and knowledge with respect to some basic properties of the human ear, as a repetition of a signal segment having a duration of less than about 50 ms is perceived as one, and only one signal, as mentioned above. Even though the signal is actually repeated, the post-masking of the ear fails to ac-

knowledge two separate signals. It should, of course, be noted that a single, full covering signal model of the ear has never been established. Nevertheless, the time constant with respect to certain types of post-masking has been experimentally determined to be dependent on the types of signals involved at about 40-60 ms. It is evident that the duration of the divided signal segment shall be so short that the ear system will not be able to perceive two repeated signals.

This feature is of particular importance when speaking about sound signals, which can only be analyzed by means of a subjective analysis performed by a listener. Fields in which the invention will provide important support include evaluation of sound signals emitted from the heart beats. A trained listener, such as a pathologist, will thus have the possibility of getting a full expression of the actual emitted sound, even if the emitted signal is a high-speed signal, such as the one provided by the heart of a child.

It should be noted that a listener will be trained to analyze slow-speed with a corresponding low pitch according to the prior art, whereas it is possible according to the invention to gain real-time analysis experience. Thus experience gained in real-time environments obtained during years of practice will be directly exploited when analyzing sound signals reproduced and processed according to the invention. A skilled listener will thus be able to utilize the possibilities provided by the invention from day to day, which has been proved by pilot tests.

It should also be noted that the invention is not restricted to one cycle repetition, as modifications of the invention with respect to the number of repetitions may vary, dependent on the current applications.

As already stated, the repeated is established in such a way that the duration of time of substantially all the signal segments is less than 60 ms. It should be noted that, within the scope of the invention, it is to a certain degree acceptable that time segments have longer duration of time, but it should nevertheless be emphasized that the number of such time segments should be kept very low.

When, as stated in claim 2, the repeated signal segments in said output signal are modified versions of the input signal segment, an advantageous embodiment of the invention is achieved. In order to reduce the chance of any echo perception further, a modification of the repeated signal segments is performed. This modification seems to increase the effect that the listener will not perceive any repetition, as a modified repeated signal tends to mask the repetition.

In a simple and preferred embodiment of the invention, the signal segments should be inverted with respect to time before the repetition, thus ensuring that the repetitions will have a kind of short duration "backward" masking. It should be noted that the necessary signal processing for obtaining the inverse repetition described above is minimal.

When, as stated in claim 3, the duration of a majority of, preferably all the signal segments is less than 40

ms, preferably 30 ms, a very preferred embodiment according to the invention is achieved, as pilot tests have turned out to be very successful with respect to e.g. stethoscopes. The embodiment of the invention thus provides no perception of echo even if the signal comprises signal components in a frequency spectrum of a recorded heart sound signal of between approximately 20Hz-2kHz.

When, as stated in claim 4, the input signal is divided into signal segments having a fixed duration less than 40 ms, preferably less than 30 ms, a simple and effective embodiment of the invention is achieved. It should be emphasized that a filtering or modification of the time signals should be processed with the aim of diminishing the effects of transitions between the coupled signal segments.

When, as stated in claim 5, the input signal is divided into signal segments, where a segment is defined between the zero crossings, an advantageous embodiment of the invention is achieved, as serious high frequency generated transients between neighbouring signal segments may be avoided. Further this definition of segments makes it easy to identify the said segments.

When, as stated in claim 6, the input signal is divided into signal segments in such a way that the gradients of the neighbouring signal segments of the output signal are substantially equal, and further said neighbouring signal segments are level-compensated, another advantageous embodiment of the invention is achieved, as a smooth transition between the neighbouring signal segments is obtained.

When, as stated in claim 7, the input signal is pre-filtered by a high pass filter in such a way that further zero crossings is obtained, another advantageous embodiment is achieved, as the filtering of the low frequency signal in the input signal will tend to establish further "natural" zero crossings, as the overlying high frequency signals will be "drawn" towards zero and amplified resulting in an increase of the number of zero crossings.

It should be noted that the creation of further zero crossings in the signal is established with the purpose of creating shorter signal segments, when the segment is defined between zero crossings as described above. An embodiment of such a filter could work by amplifying the sound segments having a long time duration between zero-crossings and attenuating the short segments.

When, as stated in claim 8, said high-pass filter has a zero at approximately 20 Hz and a pole at approximately 100Hz, a further advantageous embodiment is achieved.

When, as stated in claim 9, the input signal is pre-filtered by an iterative high pass filter until preferably all signal segments can be divided in a zero crossing of said input signal, a very convenient zero crossing generation is established. An iterative filter may thus be adapted to reiterate the input signal until all the desired and necessary zero crossings have been obtained.

It should be noted that, according to a preferred embodiment of the invention, modulations of the input signal segments may be performed so that a neighbouring signal segment has gradients having the same sign.

It should also be noted that the iterative filtering may, of course, be performed by digital processing means, in such a way that each iteration is followed by an expansion of the damped signal, thereby obtaining a minimum quantization noise.

When, as stated in claim 10, the output signal is post-filtered with inverse pre-filter, in such a way that the resulting filtering provides a substantially flat frequency response, an advantageous embodiment of the invention is achieved.

When, as stated in claim 11, the divided signal segments are modulated or filtered by means of a window function in such a way that the transitions between neighbouring signal segments are smoothed, a further advantageous embodiment of the invention is achieved, as disturbing transients resulting in click sounds, may be avoided.

It is evident that such a filtering window may be preferred in several applications, as a simple and effective elimination or damping of "transition noise" may be obtained.

When each signal segment is established in such a way that the duration of time of a majority of, preferably all the signal segments is adapted with respect to the post-masking effect in an ear in such a way that a majority of the repeated signal segments, preferably all in an audio representation of the output signal will be recognized as one segment, an optimum processing method according to the invention is provided.

BRIEF DESCRIPTION OF THE DRAWINGS

The present invention is illustrated by way of example and not limitation in the figures in which:

Figure 1 illustrates the basic parts of an electronic stethoscope.

Figure 2 is the half rate algorithm consisting of a filter algorithm, a CAS algorithm and the buffers.

Figure 3a shows the iterative filter process of the filter algorithm and figure 3b, 3c, 3d and 3e illustrates the algorithm performed on heart stepwise.

Figure 4 illustrates a signal with zero crossings and a window function.

Figure 5 illustrates the CAS algorithm when performed on a sample signal of two cycles.

Figure 6 illustrates a heart signal before pre-filtering.

Figure 7 illustrates a heart signal after pre-filtering.

Figure 8 illustrates a half rate heart signal before post-filtering.

Figure 9 illustrates a half rate heart signal after post-filtering.

Figure 10a illustrates the pre-filter and figure 10b illustrates the post-filter.

DETAILED DESCRIPTION

A preferred embodiment of the invention will be described below.

Fig 1 illustrates an electronic stethoscope consisting of a microphone 11 connected to an analog to digital converter 12 from which the output is connected to a memory and central processing unit 13. The memory and central processing unit 13 is connected to a digital to analog converter 14 and the output is connected to a speaker 15.

In use, the physician places the microphone 11, which may be in the shape of a bell, on the patient's chest and the sound is recorded and processed in the processing unit 13. It is possible to hear the processed signal by using the speaker 15 connected to the digital to analog converter 14.

Fig 2 illustrates the half rate algorithm performed by the central processing unit and memory 13. This algorithm consists of two parts - a filter algorithm 21 and a CAS (Copy And Splice) algorithm 22. The recorded data is placed in the input work buffer 23, and then filtered using an iterative filter 24 and 24'. The CAS algorithm 22 is performed between the filters 24 and 24'. The CAS algorithm consists of a zero crossing locator 25, a window function 26 and a copy and splice function 27. The algorithm halves the rate of the sound, resulting in a doubling of the length of the sound signal, whereby the signal has twice the original duration in the output work buffer 28.

The iterative filter algorithm including the pre-filter 35 and the post-filter 37 is shown in figure 3a. The in-

put signal is pre-filtered. This is done in order to amplify the high frequency signal elements and attenuate the lower frequencies. In order to reduce the processor power needed by the algorithm, the algorithm is performed on the sound signal part by part. In a preferred embodiment it is run on parts with a 10 second duration. First, the time period 32 is singled out of the recorded sound signal 31, secondly an algorithm 33 determines the maximum cycle time. In 34 the algorithm checks whether the cycle time is above a predetermined value T_{max}. If it is above the predetermined value, the signal part is filtered using a high pass filter 35, and this step is repeated until the cycle time is below the predetermined value. Then, the CAS algorithm is executed in 36, in the described embodiment the algorithm doubles the length of the sound signal. Finally, the signal part is post-filtered the same number of times as it was pre-filtered using a low pass filter 37 which has an inverse transfer function with respect to the pre-filter 35. The post-filtering 37 amplifies the low frequencies (long cycles) in the same way as they were attenuated in the pre-filter 35 in order to ensure a flat frequency response from input to output. To avoid echo, the value T_{max} should be chosen according to the time constant of the ear, which is the response time for the human ear after hearing a first sound.

To illustrate the effect of the iterative filter, a signal is shown before filtering in figure 3b. Then the signal is shown after one filtering in figure 3c, followed by the signal after filtering twice. Finally, the signal is shown after being filtered three times resulting in an extra zero crossing in the time interval between 0,6s and 0.65s.

The CAS algorithm shown in figure 2 will be described in detail below. A sample signal 43 is shown in figure 4. The zero crossing locator 25 in figure 2 locates the negative to positive transitions 41 (zero crossings) in the filtered input signal. This means that the boundaries of all cycles in the signal are located. These locations will be used by the window function 26 shown in figure 2. The window function 26 is used to prevent click sounds from occurring when succeeding cycles are pasted together, the start and end portion of each cycle are smoothed (faded in/out). The window 42 will generate signal portions that are a bit longer than those of the cycle itself (zero crossing to zero crossing), this is done to enable smooth overlapping sections, from one cycle to the next. In the preferred embodiment the amplitude (weight) of the window 42 at its centre equals 1.0, and the weight at the zero crossings 41 equals 0.5. This results in the cycle after cutting 44 being a bit longer than from zero crossing to zero crossing, providing smooth transitions between succeeding cycles.

An example of how the output signal of a half-rate signal is made by using the copy and splice process 27 from figure 2 is shown in figure 5. A sample input signal after pre-filtering 51 consists of two cycles 52 and 53. The cycle 52 is cut from the sample signal 51, using the window 42 shown in figure 4, providing the signal 54. It is seen that the window described above used on the cycle 52 results in signal 54 with longer duration than the identified cycle 52. The signal 54 is then copied and shifted in time resulting in the signal 55, which succeeds the signal 54 with overlapping zero crossing. In a preferred embodiment as shown in figure 5, the copy 55 is mirrored

in both the horizontal axis and the vertical. Tests have shown that this results in a minimum chance of echo perception. The similar is done to the cycle 53, first the cycle is cut from the signal 51 using the window 42 providing the signal 56. Then the signal 56 is copied and mirrored providing the signal 57. The signals 54, 55, 56 and 57 are added causing a reduction of the rate of the signal 51 by 50%. The original pitch is obtained, and performing this process on only fast cycles, the listener will not experience any echoes.

This method used on a heart signal is shown in the following.

Figure 6 shows a heart signal before filtering with the iterative high-pass pre-filter 24 from figure 2. It is obvious that the signal includes some slow cycles.

Figure 7 shows the signal after filtering with the high pass filter 24, and it is obvious that the slow cycles have been attenuated resulting in only fast cycles.

Figure 8 shows the pre-filtered heart signal as it is before post-filtering. The length of the signal has been doubled using the CAS algorithm.

Figure 9 shows the output signal after post-filtering. The signal has been post-filtered the same number of times as it was pre-filtered. The rate of the original sound signal has now been halved and the physician listening to this halved version will not be able to perceive any echo.

Figure 10 shows the pre-filter and the post-filter, which in combination have a flat frequency response.

It should be noted that in this preferred embodiment of the invention the rate of the signal has been halved but it is also possible to reduce the speed by another fraction. This reduction depends on how many times the cycles are repeated. In this embodiment cycles were used as the segments to be copied, but other methods could also be used to define a segment. Though it is advantageously to use segments that makes it possible to get a smooth transition between neighboring segments.

CLAIMS

1. A method of processing a signal representing an input sound signal, said signal being divided into a plurality of signal segments each having an individual duration of time, said signal segments being processed into an output signal of successive signal segments, said signal segments being processed in such a way that at least one, preferably all signal segments are repeated immediately and successively at least once in said output signal

characterized in that

each signal segment is established in such a way that the duration of time of substantially all the signal segments is less than 60 ms.

2. A method of processing a signal representing an input sound signal according to claim 1, wherein the repeated signal segments in said output signal are modified versions of the input signal segment.

3. A method of processing a signal according to claims 1 and 2, wherein the duration of the majority of, preferably all the signal segments is less than 40 ms, preferably 30 ms.

4. A method of processing a signal according to claims 1 - 3, wherein the input signal is divided into signal segments having a fixed duration of time of less than 40 ms, preferably less than 30 ms.

5. A method of processing a signal, according to claims 1 - 4, wherein the input signal is divided into signal segments in zero crossings.

6. A method of processing a signal according to claims 1 - 5, wherein the input signal is divided into signal segments in such a way that the gradients of the neighboring signal segments of the output signal are almost equal, wherein said neighboring signal segments are level-compensated.

7. A method of processing a signal according to claims 1 - 6, wherein the input signal is pre-filtered by a high-pass filter in such a way that further zero crossings may be obtained.

8. A method of processing a signal according to claims 1 - 7, wherein said high pass filter has a zero at approximately 20 Hz and a pole at approximately 100 Hz.

9. A method of processing a signal according to claims 1 - 8, wherein the input signal is pre-filtered by an iterative high-pass filter until preferably all signal segments are divided in a zero crossing of said input signal.

10. A method of processing a signal according to claim 9, wherein the output signal is post-filtered with an inverse pre-filter in such a way that a resulting filtering provides a substantially flat frequency response.

11. A method of processing a signal according to claims 1-10, wherein the signal divided segments are multiplied or filtered by means of a window function in such a way

that the transitions between neighbouring signal segments are smoothed.

12. A method of processing a signal representing an input sound signal, said signal being divided into a plurality of signal segments each having an individual duration of time, said signal segments being processed into an output signal of successive signal segments, said signal segments being processed in such a way that at least one~~,~~ preferably all signal segments are repeated immediately and successively at least once in said output signal

characterized in that

each signal segment is established in such a way that the duration of time of a majority of, preferably all the signal segments is adapted with respect to the post-masking effect in the ear in such a way that a majority of the repeated signal segments, preferably all in an audio representation of the output signal will be recognized as one segment.

13. An apparatus comprising digital processing means, said apparatus being adapted to perform the method of claims 1-12.

14. An apparatus for processing a signal representing an input sound signal, said apparatus comprising signal processing means for dividing said input signal into a plurality of signal segments each having an individual duration of time, said signal segments being processed into an output signal of successive signal segments, said signal segments being processed in such a way that at least one, preferably all signal segments are repeated

immediately and successively at least once in said output signal

characterized in that

said apparatus further comprises means for establishing each signal segment in such a way that substantially all the signal segments are less than 60 ms.

15. An apparatus according to claim 14 wherein the repeated signal segments in said output signal are modified versions of the input signal segment.

16. An apparatus according to claims 14 and 15, wherein the duration of the majority of, preferably all the signal segments are less than 40 ms, preferably 30 ms.

17. An apparatus according to claims 14 - 16, wherein the input signal is divided into periods of signal segments in zero crossings.

18. An apparatus according to claims 14 - 17, wherein the input signal is pre-filtered by a high pass filter in such a way that further zero crossings may be obtained.

19. An apparatus according to claims 14 - 18, wherein the input signal is pre-filtered by an iterative high-pass filter until preferably all signal segments are divided into a zero crossing of said input signal.

20. An apparatus according to claims 14 - 19 wherein the output signal is post-filtered with an inverse pre-filter in such a way that a resulting filtering provides a substantially flat frequency response.

21. An apparatus according to claims 14-20, wherein the signal divided segments are multiplied or filtered by means of a window function in such a way that the transitions between neighbouring signal segments are smoothed.

22. An electronic stethoscope comprising at least one input transducer and at least one output transducer, said stethoscope comprising signal processing means for dividing said input signal in time into a plurality of signal segments each having an individual duration of time, said signal segments being processed into an output signal of successive signal segments, said signal segments being processed in such a way that at least one, preferably all signal segments are repeated immediately and successively at least once in said output signal

characterized in that

said stethoscope further comprises means for establishing each signal segment in such a way that the duration of time of substantially all the signal segments is less than 60 ms and said stethoscope further comprising means for reproducing said output signal by means of said output transducers.

ABSTRACT

The invention relates to a method of processing a signal representing an input sound signal, said signal being divided in time into a plurality of signal segments each having an individual duration of time, said signal segments being processed into an output signal of successive signal segments in such a way that at least one, preferably all signal segments are repeated immediately and successively at least once in said output signal, wherein each signal segment is established in such a way that the duration of time of a majority of, preferably all the signal segments is less than 60 ms.
Thus, according to the invention, a sound signal can be reduced in speed, by doubling the number of short cycles. The method has special interest in the field of auscultation, and specially the auscultation of heart beats. By using fast cycles the physician will not be able to perceive echoes, which is the problem with prior art. The invention provides a method for reducing the speed of a sound signal, resulting in a sound signal reduced in time.

(Fig.2)

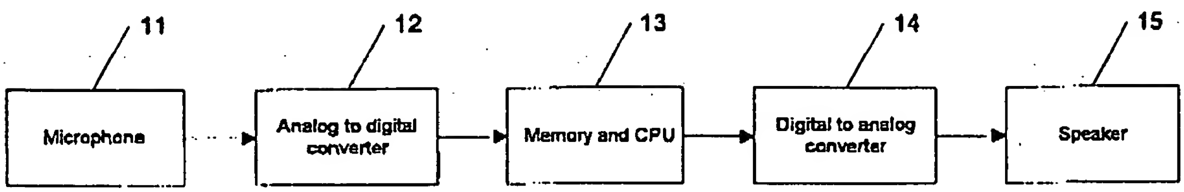


Figure 1

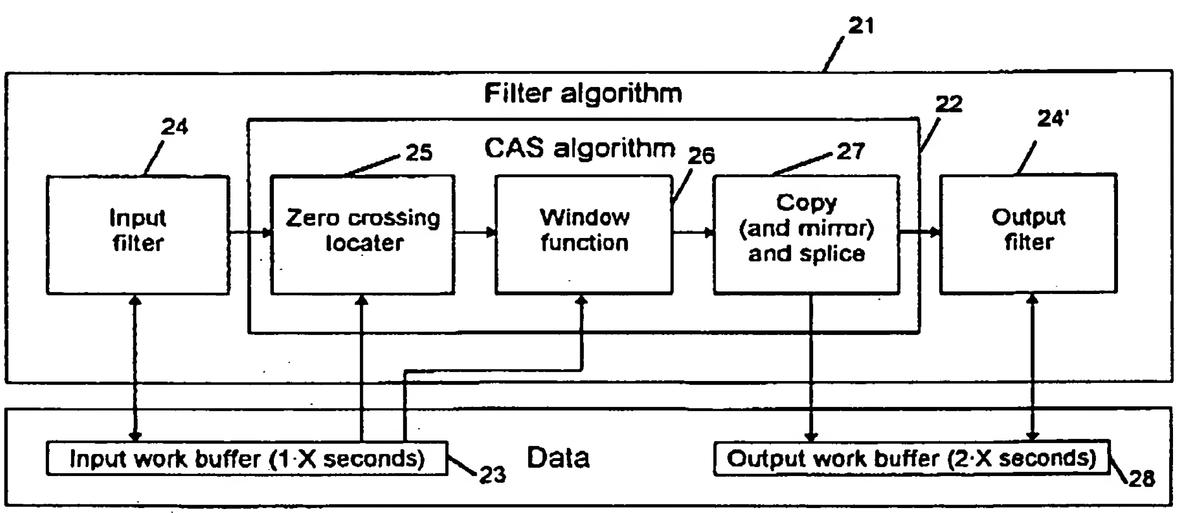
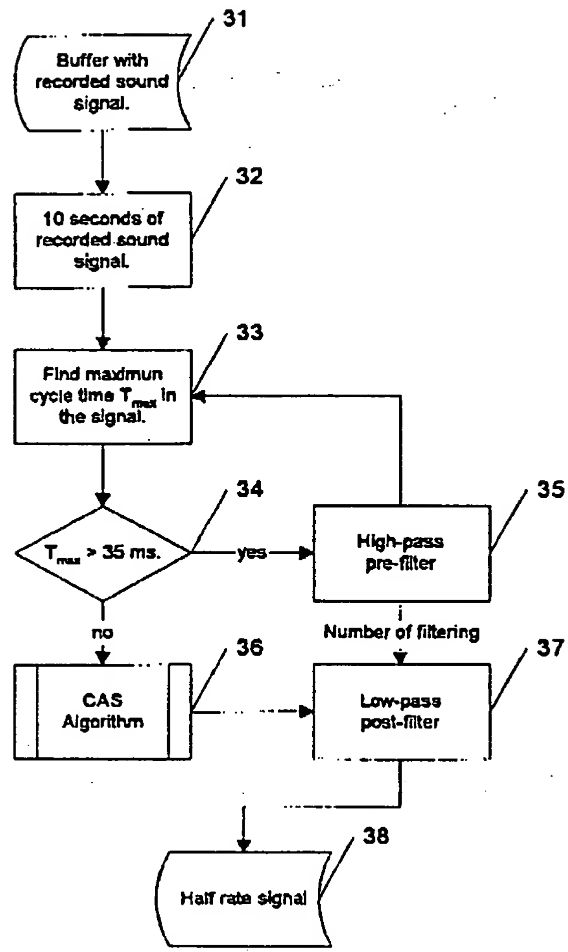


Figure 2



31
Buffer with recorded sound signal.
32
10 seconds of recorded sound signal.
33
Find maximun cycle time T_{max} in the signal.
34
T_{max} > 35 ms.
yes
35
High-pass pre-filter
Number of filtering
no
36
CAS Algorithm
37
Low-pass post-filter
38
Half rate signal

Figure 3a

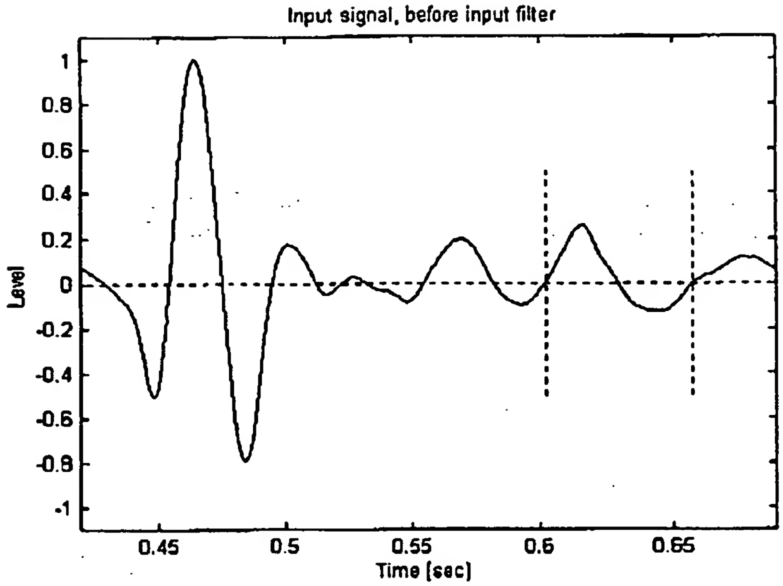


Figure 3b

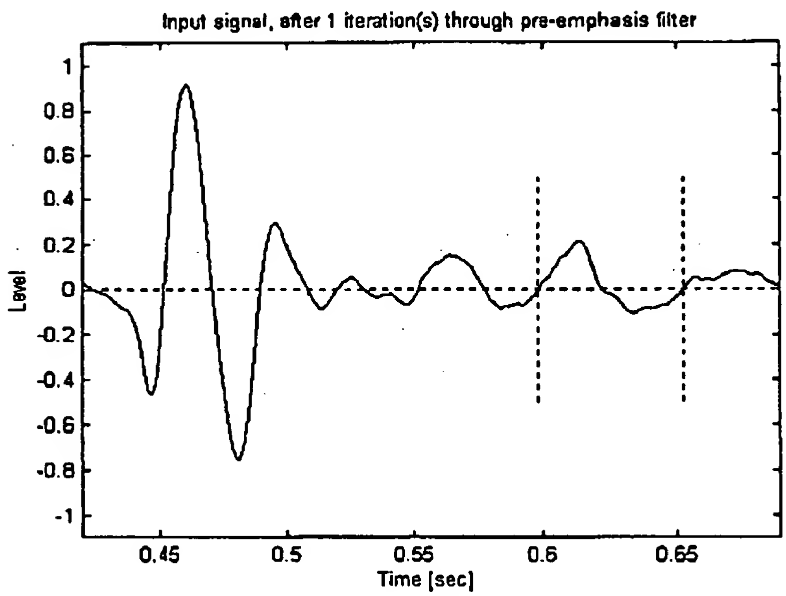


Figure 3c

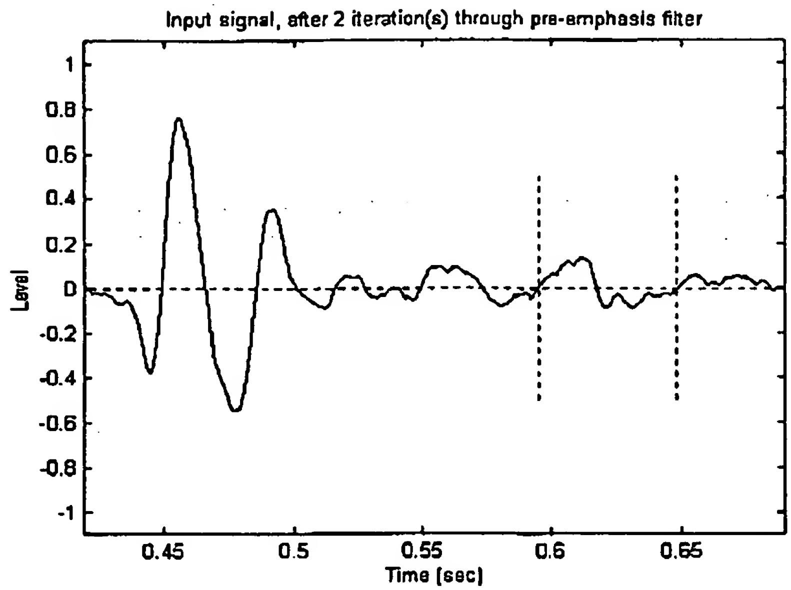


Figure 3d

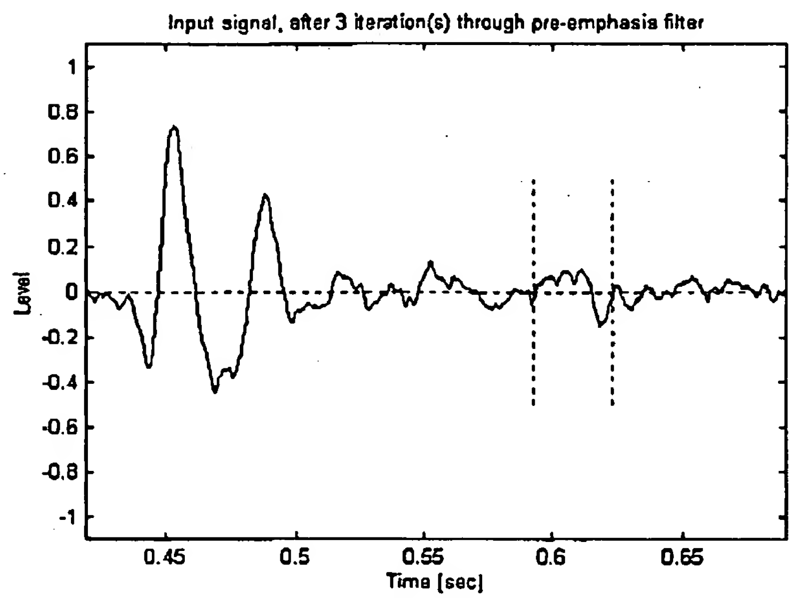


Figure 3e

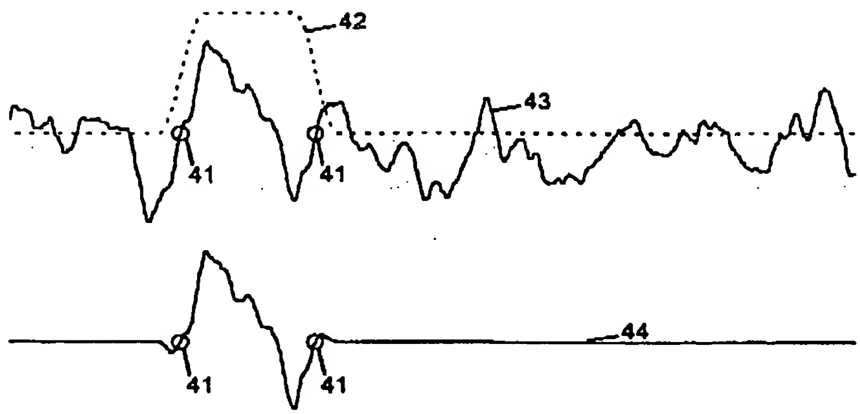
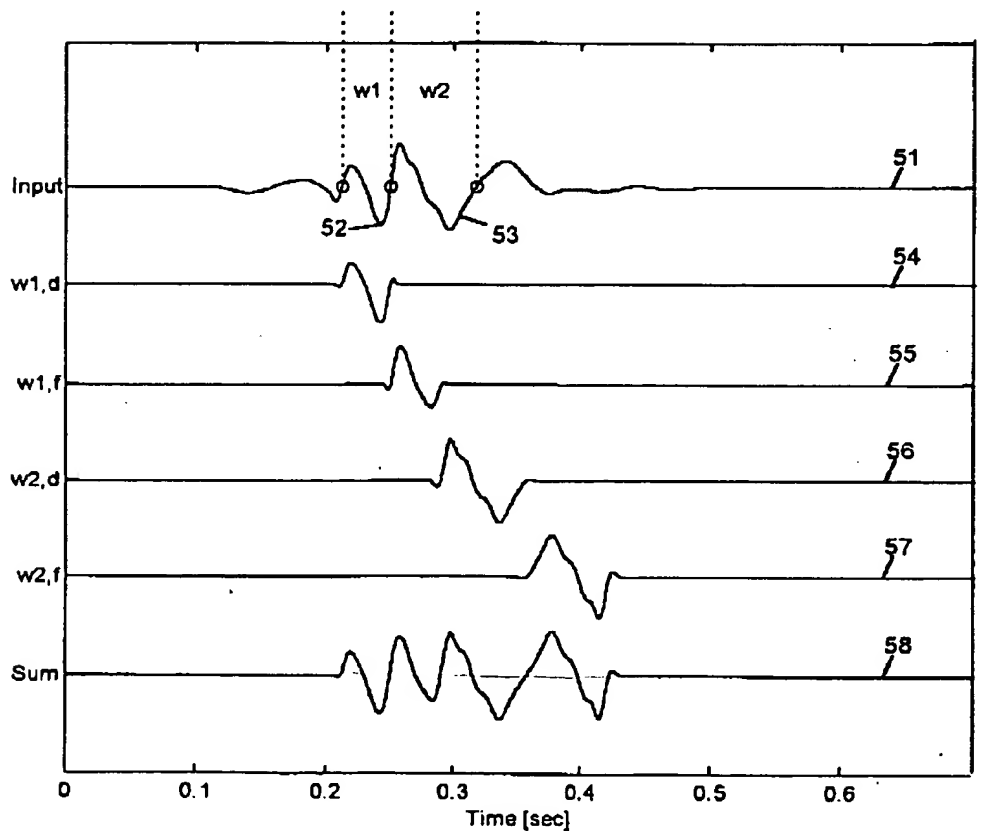


Figure 4



Σ54,55...

Figure 5

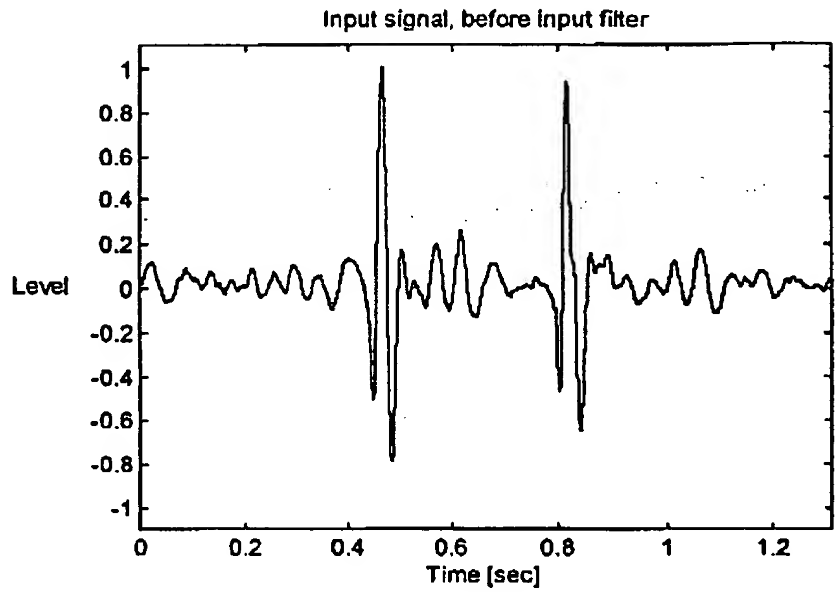


Figure 6

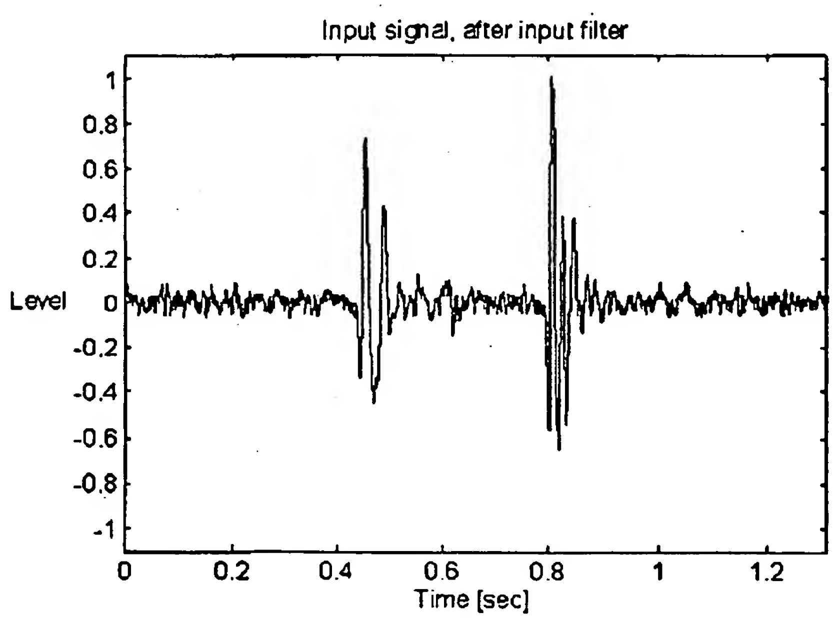


Figure 7

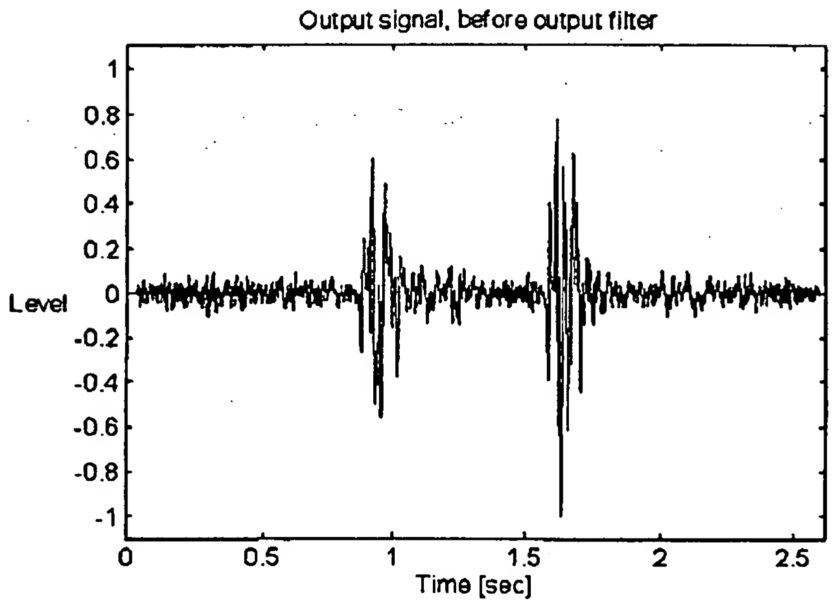


Figure 8

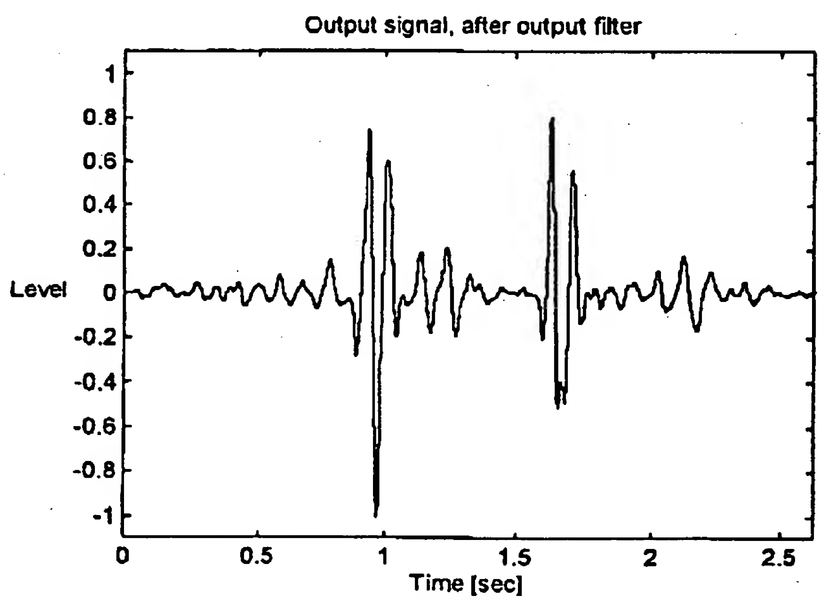


Figure 9

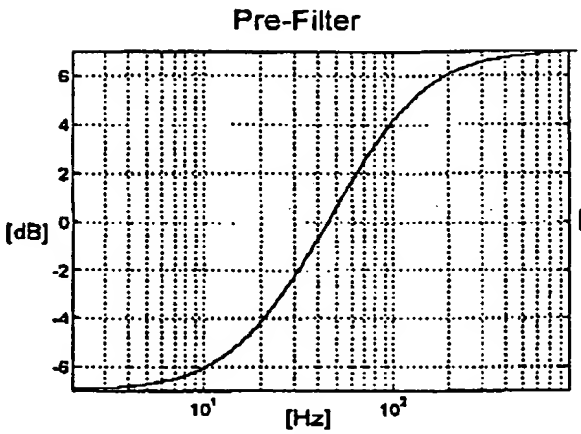


Figure 10a

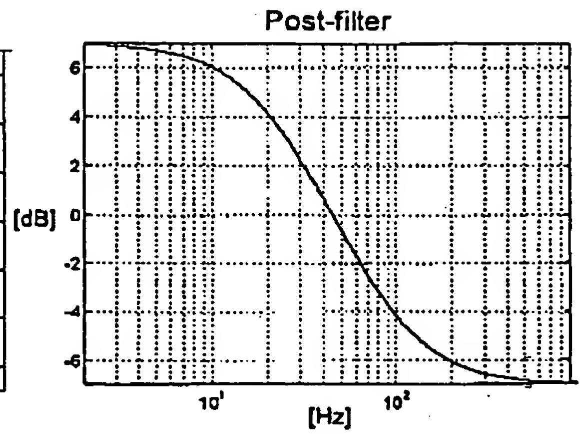


Figure 10b